> "Transforming input signal to a time-frequency representation is taught by Liu;
> Estimating background noise is taught by Liu;
> Comparing time-frequency representation with a signal node is taught by Liu;
> Determining a template in the signal is taught by Liu; and
> Replacing the acoustic input signal is taught my Liu.
>
> Liu does not specifically teach replacing the acoustic input signal with a low-noise output signal comprising a mix of the input signal and best matching template. However, refer to Miseki et al. who teach a method for encoding and decoding a speech signal including background noise wherein an input signal is separated into a speech component and a background noise component(isolating sounds) and a multiplexer multiplexes the data of the two components to produce a low noise input signal(abstract), for the purpose of efficiently encoding and decoding a speech signal which includes background noise such that the compressed speech is as close to the original speech as possible (Miseki et al. at col 1, lines 6-11) "

Applicant respectfully disagrees and submits that claims 1, 3, and 5 are in a condition for allowance and argues as follows:

I. Liu in view of Miseki et al. does not teach, suggest or describe replacing a digitized acoustic input signal with a low-noise output signal comprising a mix of digitized acoustic input signal and a best matching template.

Claims 1, 3, and 5 include the following limitation

"... replacing the digitized acoustic input signal with a low-noise output signal comprising a mix of the digitized acoustic input signal and the best matching template."

Liu only considers best matching template as an input signal (see Liu at col 10, lines 20-23). Liu does not teach or suggest that the input signal is mixed with the templates. For example, column 10, lines 10-12 of Liu states:

> "... If the codebook is designed in a quiet background while the input speech comes from a noisy environment, selection of the optimum word becomes difficult. "

and further at column 10, lines 16-23 Liu states:

> "... To overcome this drawback, adaptive vector quantization is used in the present invention. This refers to the updating of the original codebook C based upon an estimate of the background noise level to generate a "noisy" codebook C'. The noisy codebook C' is searched to find the best match with the input vector, then the index for the corresponding clean codeword is selected for transmission, and is also used at the receiver end for synthesis"

As Liu clearly teach that the signal which is transmitted and processed for synthesis is only the best match template and not a mix of input signal and a best matching template. Hence Liu does not teach, suggest or describe replacing a digitized acoustic input signal with low-noise output signal comprising a mix of digitized acoustic input signal and a best matching template.

At column 2, lines 28-35 Miseki et al. states as following:

> "... According to the present invention, an input speech signal is separated into a first component mainly constituted by speech and a second component mainly constituted by a background noise at each predetermined unit of time, and encoding is performed using methods for encoding based on different models which are respectively adapted to the characteristics of the speech and background noise to improve the efficiency of the encoding as a whole."

Even if Miseki et al. do teach encoding speech signal and background noise separately, it is with respect to efficient encoding of noisy speech(see Miseki et al. at col 1, line 6-11). Miseki et al. does not teach, suggest or describe replacing a digitized acoustic input signal with a low-noise output signal comprising a mix of digitized acoustic input signal and a best matching template.

Therefore, Liu in view of Miseki et al. does not teach, suggest or describe the present invention of claims 1, 3, and 5.

II Rejection of Claims 2, 4 and 6 Based on 35 U.S.C § 103(a)

The Examiner has rejected claims 2, 4 and 6 based on 35 U.S.C § 103(a) as being unpatentable over Liu (US Patent No: 5,680,508) in view of Fink et al. (US Patent No: 6,167,375).

The Examiner has rejected claims 2, 4, and 6 over Liu as follows:

> "Transforming input signal to a time-frequency representation is taught Liu;
> Comparing time-frequency representation with a signal model is taught Liu;
> Determining a template in the signal model is taught Liu;
> Synthesizing a signal based on the best matching template is taught Liu; and
> Isolating transient sound and including long transients without signal content and background noise is taught Liu.
>
> Liu does not specifically teach isolating transient sounds and including transients in the estimation of the background noise. however, refer to Fink et al, who teach a method of transforming a speech signal with separates a speech signal into two signal parts(which includes a transient portion) and suggest implementation of the method as being extremely expedient for synthesizing well-defined sounds "

Applicant respectfully disagrees and submits that claims 2, 4, and 6 are in a condition for allowance and argues as follows:

Claims 2, 4 and 6 include the limitation

> "... .resynthesizing a low-noise output signal using the best matching template."

I Liu in view of Fink et al. does not teach, suggest or describe resynthesizing a low-noise output signal using the best matching template

Liu only considers best matching template as an input signal. This best matched template is then selected for transmission and also used at receiver end for synthesis. For example Liu at column 10 line 20-23 states as following:

> "The noisy codebook C' is searched to find the best match with the input vector, then the index for the corresponding clean codeword is selected for transmission, and is also used at the receiver end for synthesis"

Even if Liu teach synthesizing of an input signal, it is only the best match template which is being synthesized. Liu does not teach, suggest or describe using a low-noise output signal, which is

a mix of digitized acoustic signal and the best matching template, to be resythesized using the best matching template.

Even if Fink et al. teach separating speech signal into two parts of which one include transient portion(see Frink et al. at Abstract), it fails to teach, suggest or describe resynthesizing a low-noise output signal using the best matching template. For example, Fink et al. at column 2, lines 21-35 states:

> "... This object is obtained by a method of transforming a speech signal, comprising separating the speech signal into two signal parts (a, b) where 'a' represents the quasistationary part of the signal with information on the formant frequencies, and 'b' represents a residual signal with the transient part of the signal containing information on pitch-frequency and stop consonants, said signal b being produced by inverse filtration of the speech signal, characterized in that, after the inverse filtration, the signal b is supplied in parallel to a transient detector and a pitch manipulator comprising a delay circuit which is serially coupled to a multiplier to which the output signal is supplied from the transient detector."

and further at column 3 lines 45-51, Fink et al. states as follows:

> "The invention also concerns an apparatus for transforming a speech signal, comprising a circuit for splitting the signal into two parts a, b where the first part is supplied to a decomposition circuit in series with a transformation circuit, and the other b is supplied to a circuit for inverse filtration. This apparatus is characterized in that the output from the circuit is connected in parallel to a transient detector and a pitch manipulator comprising a series connection of delay circuit and a multiplier circuit to which the output from the transient detector is connected."

Fink et al. teaches separating the signal into two parts and then manipulating these two parts using multipliers independently of each other, hence changing the frequency of the signal without changing the information. Fink et al. does not teach, suggest or describe resynthesizing a low-noise output signal using the best matching template.

Therefore, Liu in view of Fink et al. does not teach, suggest or describe the present invention of claims 2, 4 and 6.

III Rejection of Claims 7-9 Based on 35 U.S.C § 103(a)

The Examiner has rejected claims 7 through 9 based on 35 U.S.C § 103(a) as being unpatentable over Liu (US Patent No: 5,680,508) in view of Fink et al. (US Patent No: 6,167,375) and Miseki et al. (US Patent No: 6,167,375) .

The Examiner has rejected claims 7, 8, and 9 over Liu as follows:

> "Transforming input signal to a time-frequency representation is taught by Liu;
> Comparing time-frequency representation with signal model is taught by Liu;
> Determining a template in the signal model is taught by Liu;
> Isolating transient sound and including long transients without signal content and background noise is taught Liu.
>
> Liu does not specifically teach isolating transient sounds and including transients in the estimation of the background noise. However, refer Fink et al. who teach a method of transforming a speech signal into two signal part(which includes a transient portion) and suggest implementation of method as being extremely expedient for synthesizing well-defined sounds(abstract).
>
> Replacing the acoustic input signal is taught my Liu. Liu does not specifically teach replacing the acoustic input signal with a low-noise output signal comprising a mix of the input signal and best matching template. However, refer to Miseki et al. who teach a method for encoding and decoding a speech signal including background noise wherein an input signal is separated into a speech component and a background noise component(isolating sounds) and a multiplexer multiplexes the data of the two components to produce a low noise input signal(abstract), for the purpose of efficiently encoding and decoding a speech signal which includes background noise such that the compressed speech is as close to the original speech as possible(Miseki et al. at col 1, line 6-11)"

Applicant respectfully disagrees and submits that claims 7 through 9 are in a condition for allowance and argues as follows:

Claims 7 through 9 include the following limitation

"... replacing the digitized acoustic input signal with a low-noise output signal comprising a mix of the digitized acoustic input signal and the best matching template."

I Liu in view of Miseki et al. and further in view of Fink et al does not teach, suggest or describe replacing a digitized acoustic input signal with a low-noise output signal comprising a mix of digitized acoustic input signal and a best matching template.

Liu only considers best matching template as an input signal(see Liu at col 10, lines 20-23). Liu does not teach or suggest that the input signal is mixed with the templates. For example, column 10, lines 10-12 of Liu states:

> "... If the codebook is designed in a quiet background while the input speech comes from a noisy environment, selection of the optimum word becomes difficult. "

and further at column 10 line 16-23 Liu states:

> "... To overcome this drawback, adaptive vector quantization is used in the present invention. This refers to the updating of the original codebook C based upon an estimate of the background noise level to generate a "noisy" codebook C'. The noisy codebook C' is searched to find the best match with the input vector, then the index for the corresponding clean codeword is selected for transmission, and is also used at the receiver end for synthesis"

As Liu clearly teach that the signal which is transmitted and processed for synthesis is only the best match template and not a mix of input signal and a best matching template. Hence Liu does not teach, suggest or describe replacing a digitized acoustic input signal with low-noise output signal comprising a mix of digitized acoustic input signal and a best matching template.

At column 2, lines 28-35 Miseki et al. states as following:

> "... According to the present invention, an input speech signal is separated into a first component mainly constituted by speech and a second component mainly constituted by a background noise at each predetermined unit of time, and encoding is performed using methods for encoding based on different models which are respectively adapted to the characteristics of the speech and background noise to improve the efficiency of the encoding as a whole."

Even if Miseki et al. do teach encoding speech signal and background noise separately, it is with respect to efficient encoding of noisy speech(see Miseki et al. at col 1, line 6-11). Miseki et al. does not teach, suggest or describe replacing a digitized acoustic input signal with a low-noise output signal comprising a mix of digitized acoustic input signal and a best matching template.

Fink et al. teach to separate input signal into two parts - transient portion and quasistationary (see Frink et al. at Abstract), but fails to teach, suggest or describe replacing a digitized acoustic input signal with a low-noise output signal comprising a mix of digitized acoustic input signal and a best matching template. For example, at column 2, lines 21-35 of Fink et al. states:

> "... This object is obtained by a method of transforming a speech signal, comprising separating the speech signal into two signal parts (a, b) where 'a' represents the quasistationary part of the signal with information on the formant frequencies, and 'b' represents a residual signal with the transient part of the signal containing information on pitch-frequency and stop consonants, said signal b being produced by inverse filtration of the speech signal, characterized in that, after the inverse filtration, the signal b is supplied in parallel to a transient detector and a pitch manipulator comprising a delay circuit which is serially coupled to a multiplier to which the output signal is supplied from the transient detector."

Further, Fink et al. teach completely separating low-noise signal output from best match template. For example, Fink et al. at col 2, lines 34-42 states as follows:

> "Signal pulses are captured in this manner by the transient detector, and since the signal to the multiplier is delayed with respect to the signal arriving from the transient detector, it is possible to eliminate the noise pulse by means of the multiplier. Further, it is extremely essential that the elimination of the noise pulse can take place completely independently of the signal processing in the other signal part, which comprises manipulation of the formant frequencies"

Therefore, Liu in view of Miseki et al. and further in view of Fink et al. does not teach, suggest or describe the present invention of claims 7 through 9.

CONCLUSION

For at least foregoing reasons, Applicant submits that the cited art does not teach or suggest, let alone anticipate, claims 1 through 9 of the present application. In view of above, it is submitted that the claims now in the application, i.e., claim 1 through 9 are in condition for allowance. Accordingly, reconsideration and allowance of claims 1 through 9 are requested.

Respectfully submitted,

COUDERT BROTHERS

Date: 12/13/02

By:

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